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(54) Title: METHOD AND ARRANGEMENT FOR FILTER BANK BASED SIGNAL PROCESSING
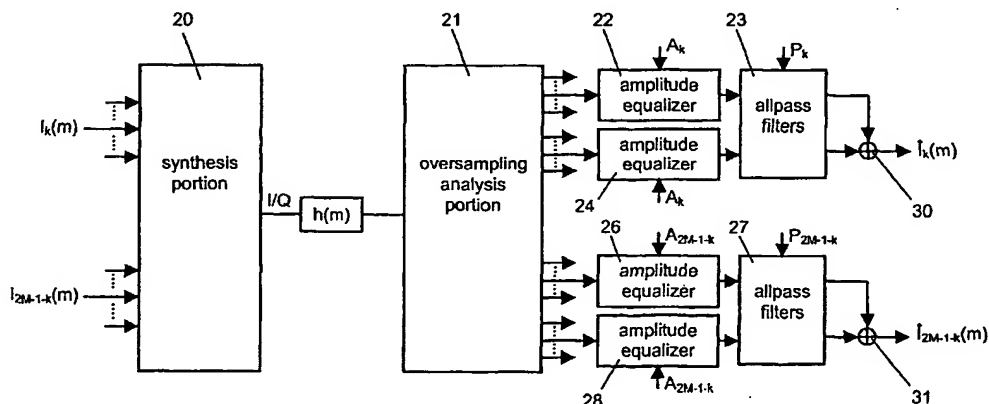
(57) Abstract: The invention relates to a method for a filter bank based signal processing system. In order to enable a signal processing with a low complexity and at the same time a good performance, a method is proposed which comprises in a first step performing a filter-bank based analysis for converting a complex higher-rate channel signal into oversampled lower-rate sub-channel signals, each sub-channel corresponding to a different frequency range. In a second step, the proposed method comprises processing the oversampled lower-rate sub-channel signals with a polynomial model of a system frequency response within the frequency range of the respective sub-channel. The invention relates equally to a unit and a system comprising means for realizing the proposed method.

METHOD AND ARRANGEMENT FOR FILTER BANK BASED SIGNAL PROCESSING

5   FIELD OF THE INVENTION

The invention relates to a method for a filter bank based
signal processing system. The invention relates equally
to a unit performing a signal processing in a filter bank
10 based signal processing system and to a filter bank based
signal processing system comprising such a unit.

BACKGROUND OF THE INVENTION

15 Processing signals comprises in a variety of systems a
channel equalization. A channel equalization is employed
for compensating the effects of a fading multipath
channel, which constitute a fundamental problem in
communication systems.
20

Various channel equalization techniques have been
developed for the traditional single-carrier transmission
systems and more recent CDMA systems. With increasing
data rates and signal bandwidths in new and future
25 systems, there is moreover an increasing interest in
multicarrier transmission techniques, for which dedicated
channel equalization techniques have to be employed. In a
multicarrier transmission system, a transmitted higher-
rate data stream is divided into a number of lower-rate
30 sub-channels partly overlapping in the frequency domain.
For multiplexing and demultiplexing these sub-channels,
various techniques are known, for instance orthogonal
Frequency Division Multiplexing (OFDM) techniques and
filter bank based multicarrier (FBMC) techniques. FBMC

techniques are sometimes also referred to as Discrete
Wavelet Multitone (DWMT) techniques.

OFDM has been described for example by R. van Nee and R.
5   Prasad in chapter 2 "OFDM basics" of the document "OFDM
Wireless Multimedia Communications", Artech House,
London, 2000. In an OFDM system and its baseband version
Discrete Multitone (DMT), a high-rate datastream is split
into a number of lower rate streams that are transmitted
10   simultaneously over a number of sub-carriers, in order to
decrease the relative amount of dispersion in time caused
by multipath delay spread. The sub-channels are
multiplexed and demultiplexed by means of an IFFT-FFT
(Inverse Fast Fourier Transform / Fast Fourier Transform)
15   pair. In OFDM and DMT systems, a time-domain guard
interval introduced for every OFDM symbol and a simple 1-
tap frequency domain equalization is commonly used for
channel equalization. In the guard time, the OFDM symbol
is cyclically extended to avoid intercarrier
20   interference.

OFDM and DMT systems are very robust from a channel
equalization point of view. On the other hand, there are
certain advantages that can be obtained by using an FBMC
25   system instead of an IFFT-FFT pair, as will be explained
in the following.

An FBMC system has been presented for example by T.
Ihalainen, Tobias Hidalgo-Stitz and Markku Renfors in:
30   "On the performance of low-complexity ASCET-equalizer for
a complex transmultiplexer in wireless mobile channel" in
Proc. 7th Int. OFDM-Workshop 2002, Harburg, Germany, pp.
122-126, Sep. 2002, which is incorporated by reference
herein.

Figure 1 is a block diagram of a 0th order ASCET (Adaptive
sine-modulated/cosine-modulated filter bank equalizers
for transmultiplexers) equalizer structure for complex
5   systems, which was taken from the above cited document
"On the performance of low-complexity ASCET-equalizer for
a complex transmultiplexer in wireless mobile channel".
The system comprises a transmitting end and a receiving
end, between which a multicarrier radio communication is
10   to be enabled.

In order to achieve a good spectral efficiency in radio
communications, it is necessary to have a complex I/Q
baseband model for the FBMC system. The equalizer
15   structure of figure 1 therefore comprises at the
transmitting end a synthesis bank for converting 2M real
low-rate sub-channel signals for transmission into a
complex I/Q (In phase / Quadrature) presentation of a
high-rate channel signal. The sampling rate conversion
20   factor is M. The synthesis filter bank includes a cosine
modulated filter bank (CMFB) 10, in which sub-filters are
formed by modulating a real low-pass prototype filter
with a cosine sequence. The cosine-modulation translates
the frequency response of the prototype filter around a
25   new center frequency. The synthesis filter bank moreover
comprises a sine modulated filter bank (SMFB) 11, in
which corresponding sub-filters are formed by modulating
a real low-pass prototype filter with a sine sequence.

30   The equalizer structure further comprises at the
receiving end an analysis bank for converting a received
high-rate channel signal into low rate sub-channel
signals again. A complex critically sampled perfect
reconstruction (PR) analysis bank would equally include a

corresponding CMFB and a corresponding SMFB, which take
the real part of the signal after the complex sub-channel
filtering. The prototype filter can be optimized in such
a manner that the filter bank satisfies the PR condition,

5   i.e. the analysis transform is invertible by the
synthesis transform. In the structure of figure 1,
however, the analysis bank implements a filter bank with
complex output signals instead of real output signals by
employing two CMFBs 12, 14 and two SMFBs 13, 15. This

10   way, oversampled sub-channel signals can be obtained for
enabling a channel equalization.

The exact equations realized by the CMFBs 10, 12, 14 and
the SMFBs 11, 13, 15 can be taken from the above cited

15   document "On the performance of low-complexity ASCET-
equalizer for a complex transmultiplexer in wireless
mobile channel".

For a transmission, 2M low-rate symbol sequences, which

20   are to be transmitted on a respective sub-channel, are
fed to the synthesis filter bank of the transmitting end,
half of them corresponding to sub-channels between 0 and
$f_s/2$, and the other half corresponding to sub-channels
between 0 and $-f_s/2$, where f_s is the high sampling rate.

25   More specifically, the difference between a respective
pair of symbols $I_k(m)$ and $I_{2M-1-k}(m)$ is divided by two and
fed to the CMFB 10, while the sum of the respective pair
of symbols $I_k(m)$ and $I_{2M-1-k}(m)$ is divided by two and fed to
the SMFB 11. In the notation $I_k(m)$ and $I_{2M-1-k}(m)$, the

30   indices indicate the respective sub-channel, while the
parameter m is a time index. The output of the SMFB 11 is
multiplied by j and then combined with the output of the
CMFB 10 in order to form a complex I/Q channel signal for
transmission. The multiplication by j means that the

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signal output by the SMFB 11 is used as the quadrature
component in the subsequent processing. The units
required for the described processing at the transmitting
end, including summing means, multiplication means, the

5   CMBF 10 and the SMBF 11, will also be referred to as
synthesis portion 20, which is indicated in figure 1 by a
first rectangle with dashed lines.

The radio channel used for transmission is equivalent to
10   a low-pass channel $H_{lp}(z)$.

At the receiving end, the high-rate channel signal is
separated again into a real part $Re\{.\}$ and an imaginary
part $Im\{.\}$, the real part $Re\{.\}$ being fed to the first
15   CMFB 12 and the first SMFB 13 of the analysis bank, and
the imaginary part $Im\{.\}$ being fed to the second CMFB 14
and the second SMFB 15 of the analysis bank. Each of the
CMFBs 12, 14 and the SMFBs 13, 15 outputs M signals via M
sub-filters.
20

Each output signal of the second SMFB 15 is subtracted
from the corresponding output signal of the first CMFB
12, resulting in a first group of signals, which
constitute an in-phase component of the first M sub-
25   channel signals. Each output of the second CMFB 14 is
added to the corresponding output of the first SMFB 13,
resulting in a second group of signals, which constitute
a quadrature component of the first M sub-channel
signals. Each output of the second CMFB 14 is subtracted
30   from the corresponding output of the first SMFB 13,
resulting in a third group of signals, which constitute a
quadrature component of the second M sub-channel signals.
Each output of the first CMFB 12 is subtracted from the
inverted corresponding output of the second SMFB 15,

resulting in a fourth group of signals, which constitute
an in-phase component of the second M sub-channel
signals. The units required for the processing at the
receiving end described so far, including separation

5   means, the CMBFs 12, 14, the SMBFs 13, 15 and summing
means, will also be referred to as analysis portion 21,
which is indicated in figure 1 by a second rectangle with
dashed lines.

10   For channel equalization, a dedicated single real
coefficient c_k, s_k, c_{2M-1-k}, s_{2M-1-k} is then used for
weighting the in-phase component and the quadrature
component of each sub-channel signal in order to adjust
the amplitude and phase of each sub-channel by a simple

15   multiplication. The indices k, 2M-1-k indicate the sub-
channel to which the respective coefficient is
associated. The coefficients c_k, s_k, c_{2M-1-k}, s_{2M-1-k} provided
for a sub-channel are preferably related to the channel
response within the corresponding sub-channel bandwidth.

20

It is mentioned in the above cited document "On the
performance of low-complexity ASCET-equalizer for a
complex transmultiplexer in wireless mobile channel" that
such a constant coefficient works well only in the case

25   when the frequency response is rather flat within each
sub-channel bandwidth, which may require a relatively
high number of sub-channels. It is further indicated that
higher-order ASCETs may be obtained by including low-
order Finite Impulse Response (FIR) filter stages for

30   each of the sub-channels. Such an approach, in which FIR
filters are used as equalizers which are adjusted using
common adaptation algorithms and criteria, like a mean-
squared error criterion, has been described for example
by B. Hirosaki in "An analysis of automatic equalizers

for orthogonally multiplexed QAM systems," IEEE Trans.
Commun., vol. 28, pp. 73-83, Jan. 1980.

The real parts of corresponding weighted signals of the
5   first and the second group of sub-channel signals are
then taken at a respective unit 16 provided to this end
and subjected to a respective decision device 18, a so
called slicer, in order to obtain the first M real sub-
channel symbol sequences $\hat{I}_k(m)$. The real parts of
10   corresponding weighted signals of the third and the
fourth group of sub-channel signals are equally taken at
a respective unit 17 provided to this end and subjected
to a respective slicer 19, in order to obtain the second
M real sub-channel symbol sequences $\hat{I}_{2N-1-k}(m)$.
15

The main characteristic of FBMC systems is that the sub-
channels can be designed optimally in the frequency
domain, e.g. to have good spectral containment. There are
certain advantages that can be obtained by using filter
20   banks with highly frequency selective sub-channels in the
transmultiplexer configuration instead of an IFFT-FFT
pair, as in the case of OFDM and DMT systems.

Firstly, the bank selectivity is a design parameter for
25   precise spectrum control. This provides resistance
against narrowband interference and allows the use of
very narrow guard bands around the multicarrier signal.
Secondly, the guard period applied in OFDM-systems to
combat intersymbol interference (ISI) becomes
30   unnecessary. Reducing the frequency-domain guard-band and
avoiding the time-domain guard interval saves significant
amount of bandwidth for data transmission, thus improving
the spectral efficiency. Furthermore, an FBMC system with
a proper channel equalization allows the use of a

considerably lower number of sub-carriers than the OFDM
techniques. This helps to reduce the problems in OFDM
which are due to a high peak-to-average power ratio.
Being able to use fewer sub-channels to cover the user
5   signal band helps to reduce the latency of the
transmission link, improves the performance in case of
time-selective channels due to a reduced symbol length,
reduces the sensitivity to Doppler effects, frequency
errors and phase noise, and gives more freedom in
10   choosing the essential system parameters.

However, the known channel equalization solutions for
FBMC systems, in which case the guard-interval approach
cannot be used, suffer from insufficient performance, as
15   in the case of the presented 0th order ASCET and/or from
relatively high implementation complexity, as in the case
of an FIR based approach.

Another structure using a filter bank system which relies
20   on an efficient sub-band processing is the analysis-
synthesis (AS) filter bank configuration. In an AS
configuration, which can be employed for various coding
and adaptive signal processing applications, the signal
frequency band is divided in an analysis bank into a
25   number of overlapping sub-bands for processing, and after
processing the signal is restored in a synthesis bank by
combining the sub-band signals again. In perfect-
reconstruction systems, the filter bank design is such
that the original signal can be restored completely, if
30   no processing is done in between. In most applications,
the system performance can be improved by increasing the
number of sub-bands. However, increasing the number of
sub-bands increases the implementation complexity, as
well as the processing latency due to the filter banks.

The use of the AS configuration in channel equalization
in single-carrier systems has been dealt with for example
by D. Falconer et al. in "Frequency domain equalization
for single-carrier broadband wireless systems", IEEE
5   Communications Magazine, vol. 40, no. 4, April 2002, pp.
58-66.

SUMMARY OF THE INVENTION

10 It is an object of the invention to enable a signal
processing in a filter bank based signal processing
system which requires a low complexity and which provides
at the same time a good performance. It is in particular
an object of the invention to enable a signal processing
15 which compensates an undesired distortion of signals in
the system.

A method for a filter bank based signal processing system
is proposed, which comprises in a first step performing a
20 filter-bank based analysis for converting a complex
higher-rate channel signal into oversampled lower-rate
sub-channel signals, each sub-channel corresponding to a
different frequency range. The proposed method comprises
in a second step processing the oversampled lower-rate
25 sub-channel signals with a polynomial model of a system
frequency response within the frequency range of the
respective sub-channel.

Further, a unit for performing a signal processing in a
30 filter bank based signal processing system is proposed.
This unit comprises an analysis filter-bank with a
plurality of sub-channel filters for converting a complex
higher-rate channel signal input to the unit into
oversampled lower-rate sub-channel signals, each sub-

channel corresponding to a different frequency range. In
addition the proposed unit comprises a filter structure
for processing oversampled lower-rate sub-channel signals
with a polynomial model of a system frequency response
5   within the frequency range of the respective sub-channel.

Finally, a filter bank based signal processing system is
proposed which comprises the proposed unit.

10 The invention proceeds from the idea that a simplified
model for the system frequency response within each sub-
channel bandwidth can be on the one hand much closer to
the real system frequency response than the piece-wise
constant frequency response model, and on the other hand
15 less complex than an accurate model for the system
frequency response. Therefore, it is proposed to use an
oversampled analysis bank and to model the relevant
spectrum or frequency response using a polynomial model
in the frequency range of each sub-band as basis for a
20 sub-channel processing.

It is an advantage of the invention that it provides a
low-complexity solution with good performance for a sub-
channel processing, e.g. a channel equalization, while
25 maintaining at the same time the advantages of sub-band
based signal processing techniques utilizing nearly or
fully perfect reconstruction filter banks.

For the special case of a channel equalization, for
30 example, the invention allows to approximate the ideal
frequency response model with good performance using a
considerably lower number of sub-bands than a 0th order
equalizer, in which amplitude and phase are assumed to be
constant within each sub-band. In comparison to other

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FBMC approaches with higher-order equalizers, the used
polynomial frequency response model reduces the
complexity and/or improves the performance of the channel
estimation block by reducing the number of parameters

5   that are to be estimated. In case of a direct adaptive
equalization, the invention moreover improves the
convergence speed. The invention thus provides in general
a better tradeoff between performance and complexity than
the known channel equalization methods for FBMC systems.

10

For realizing the oversampling filter bank analysis, the
filter bank preferably comprises sine-modulated and
cosine-modulated filter bank sections. Further
preferably, the analysis is two times oversampled and

15   provides output signals in complex I/Q format. It is to
be noted, however, that the invention can be employed for
higher oversampling factors as well.

Advantageously, the polynomial model employed for sub-

20   channel processing is a low-order polynomial model, which
comprises amplitude and phase response models of a
respective sub-band.

The polynomial model can comprise in particular a

25   linearly frequency dependent model for the amplitude
response and a linearly frequency dependent model for the
phase responses within each sub-channel frequency band.
Alternatively, other low-order polynomial models for
amplitude and phase responses can be used, for instance

30   2nd order or 3rd order polynomial models. The models can
also be piece-wise linear or low-order polynomial models
for real and imaginary parts of the system frequency
response.

The sub-channel processing can be realized for example for each sub-band with an amplitude equalizer and an all-pass filter as phase equalizer.

5   The invention can be employed as well in analysis-synthesis (AS) filter bank configurations as in synthesis-analysis filter bank configurations for transmultiplexers (TMUX).

10   In case the invention is implemented for a TMUX configuration, for example the TMUX configuration described above with reference to figure 1, it may provide a low-complexity solution for the channel equalization in FBMC systems, if the sub-channel
15   processing according to the invention forms part of the channel equalization.

AS configurations are employed for example for transform-domain adaptive signal processing techniques, like
20   adaptive equalizers, for interference cancellers or for system identification tasks. Frequency-domain equalization in single-carrier transmission systems is one particular example of interest. In general, the invention provides a better quality with a given number
25   of sub-channels than the existing approaches because the system is able to model better the ideal frequency response. Alternatively, for given performance requirements, it is possible to reduce the number of sub-bands, which helps to reduce the implementation
30   complexity, as well as the processing latency, which may become critical in many applications. The AS configuration may be employed in particular in a channel equalization in a single carrier transmission system, in which the sub-channel processing according to the

invention forms part of the channel equalization. An AS
configuration according to the invention may be used in
many other signal processing applications as well,
though.

5

The method of the invention can be realized for instance
with a signal processing algorithm, e.g. a channel
equalization algorithm. Such an algorithm can be
implemented for example as a digital VLSI (Very Large

10   Scale Integration) circuit or by using a DSP (Digital
Signal Processing) processor.

Other objects and features of the present invention will
become apparent from the following detailed description

15   considered in conjunction with the accompanying drawings.
It is to be understood, however, that the drawings are
designed solely for purposes of illustration and not as a
definition of the limits of the invention, for which
reference should be made to the appended claims. It

20   should be further understood that the drawings are not
drawn to scale and that they are merely intended to
conceptually illustrate the structures and procedures
described herein.

25   BRIEF DESCRIPTION OF THE FIGURES

      Fig. 1   is a block diagram of a known 0th order ASCET
                equalizer structure; and
      Fig. 2   is a schematic block diagram of an embodiment of
30   the system according to the invention.

DETAILED DESCRIPTION OF THE INVENTION

The system illustrated in figure 1 was already described
above. An embodiment of the system according to the
invention, which is an enhancement of the system of
figure 1, will now be described with reference to figure

5   2.

The system of figure 2 comprises a transmitter and a
receiver between which multicarrier signals are to be
transmitted via the radio interface. The system of figure

10   2 utilizes to this end a filter bank structure which is
based on sine-modulated and cosine-modulated filter bank
sections in a transmultiplexer configuration. The
equalization scheme realized in this embodiment is called
AP-ASCET (Amplitude-Phase Adaptive sine-modulated/cosine-

15   modulated filter bank equalizers for transmultiplexers).

The transmitter of the system of figure 2 includes a
synthesis portion 20 with a synthesis bank. The synthesis
bank comprises for 2M input low-rate sub-channel signals

20   a dedicated up-conversion section with a conversion
factor of M and a processing function $f_k(m)$, which
constitutes the impulse response for a sub-channel
filtering of a particular sub-channel. The index k of the
function f indicates the respective sub-channel for which

25   the function is provided, while the parameter m is a time
index. The synthesis bank may, but does not have to be
structured and operated exactly like the synthesis bank
10, 11 of figure 1.

30   The receiver of the system of figure 2 includes an
analysis portion 21 with an analysis bank. The analysis
bank comprises for each of the 2M sub-channels a cosine-
based processing function $g_c^k(m)$ followed by a down-
conversion section with a conversion factor of M,

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outputting a respective in-phase signal. The analysis
bank further comprises for each of the 2M sub-channels a
sine-based processing function $g_s^k(m)$ followed by a down-
conversion section with a conversion factor of M,

5   outputting a respective quadrature signal. The indices k
indicate again a respective sub-channel, while the
parameter m is a time index. The analysis bank in the
analysis portion 21 is implemented in the two-times
oversampled form by taking the output signals in complex

10   I/Q format. Oversampling makes it possible to perform the
channel equalization within each sub-channel
independently of the other sub-channels, since it enables
a per-carrier equalization. A typical case with 100%
roll-off, or lower, is assumed in the filter bank design

15   so that the sub-band frequency range is twice the sub-
band spacing and that two times oversampling is
sufficient to keep all unwanted aliasing signal
components below a level determined by the stopband
attenuation. The analysis bank may, but does not have to

20   be structured and operated exactly like the analysis bank
12-15 of figure 1.

In contrast to the system of figure 1, the I and Q
outputs of the analysis portion 21 of figure 2 are

25   connected for each of the sub-channels to a dedicated
special filter structure. Each filter structure comprises
a amplitude equalizer 22, 26 connected to the I output of
the analysis portion 21 for a specific sub-channel and a
amplitude equalizer 24, 28 connected to the Q output of

30   the analysis portion 21 for a specific sub-channel. Each
amplitude equalizer 22, 24, 26, 28 constitutes a three-
tap real, antisymmetric FIR filter as linear phase
amplitude correction stage. Each filter structure further
comprises an allpass filter 23, 27 functioning as a phase

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equalizer for each sub-channel. The outputs of the two
amplitude equalizers 22/24, 26/28 associated to a
respective sub-channel are connected to two inputs of the
allpass filter 23, 27 associated to this sub-channel. The
5   allpass filters 23, 27 may comprise in particular a
cascade of two complex allpass phase correction stages
and a phase rotation portion. Regardless of whether a
single allpass phase correction stage or two allpass
phase correction stages are used for each allpass filter
10   23, 27, first-order complex allpass phase correction
stages are employed in order to achieve a good
performance. The filter structure can be realized by
hardware or software. The two outputs of a respective
allpass filter 23, 27 are connected to a unit 30, 31
15   taking the real part of provided signals.

The filter structure comprises a combination of amplitude
and phase equalizers, in order to be able to compensate
Inter-Carrier- and Inter-Symbol-Interference. Non-ideal
20   channels cause phase distortions, resulting in a rotation
between real- and imaginary branches, and thus causing
Inter-Carrier-Interference, while Inter-Symbol-
Interference is caused mainly by amplitude distortion.

25   For a transmission, 2M low-rate symbol sequences $I_k(m)$,
$I_{2M-1-k}(m)$, which are to be transmitted on sub-channels k,
2M-1-k, are fed to the synthesis filter bank of the
transmitting end, half of them corresponding to sub-
channels between 0 and $f_s/2$, and the other half
30   corresponding to sub-channels between 0 and $-f_s/2$, where
f_s is the high sampling rate. In the notation $I_k(m)$,
$I_{2M-1-k}(m)$, the indices k, 2M-1-k indicate again a
respective sub-channel, while the parameter m is a time
index. The 2M sub-channel symbol sequences $I_k(m)$, $I_{2M-1-k}(m)$

are processed in the synthesis portion 20, transmitted
via the radio interface, where they undergo a channel
distortion h(m), the parameter m being again a time
index, received by the receiver and processed by the
5   analysis portion 21, e.g. as described above with
reference to figure 1. The sub-channels k and 2M-1-k,
which are located symmetrically with respect to the zero-
frequency in the baseband model, are equally located
symmetrically with respect to the radio frequency carrier
10   frequency in the modulated signals.

The analysis portion outputs for each of the 2M sub-
channels an in-phase component and a quadrature
component, e.g. like in the system of figure 1 signals of
15   a first, second, third and fourth group of low-rate sub-
channel signals. The subsequent channel equalization,
however, is not realized as in the system of figure 1
simply by multiplying the output of each sub-band filter
with a fixed complex coefficient c_k, s_k.

20

For the channel equalization in the system of figure 2, a
linearly frequency-dependent amplitude model A_k, A_{2M-1-k} is
provided to each of the amplitude equalizers 22, 24, 26,
28, and a linearly frequency-dependant phase model P_k,
25   P_{2M-1-k} is provided to each of the allpass filters 23, 27.
The respective index k, 2M-1-k of the models indicates
the sub-channel to which the filter structure is
associated and to which the respective models are
provided. It is to be noted that while separate amplitude
30   equalizer can be implemented for the I and Q branches of
a respective sub-channel by including the same real
filter in the I and Q branches, the phase equalization by
the allpass filters involves both I and Q signals, thus a
shared allpass filter is provided for the I and Q

branches of a respective sub-channel. The phase equalizer
part realized by the allpass filters includes also a
complex coefficient. Each amplitude model comprises the
value of the amplitude of the channel response at the
5   center frequency of the respective sub-channel and the
slope of the amplitude. Each phase model comprises the
value of the phase of the channel response at the center
frequency of the respective sub-channel and the slope of
the phase. Thus, four parameters which define the
10 frequency characteristics within each sub-channel are
provided to a respective filter structure.

The four parameters are provided to each filter structure
by a channel estimation block of the receiver (not
15 shown). The channel estimation block determines the
parameters based on known pilot signals transmitted in
all or some of the sub-channels from the transmitter to
the receiver. Alternatively, a so-called blind method
could be employed for determining the parameters, which
20 would not require pilot signals.

It is to be noted that while a linear frequency dependent
model is proposed here, a 2nd order model, e.g. in the
form $a_0+a_1*x+a_2*x^2$, or a 3rd order model, e.g. in the form
25 $a_0+a_1*x+a_2*x^2+a_3*x^3$, could be employed as well, where a_0,
a_1, a_2 and a_3 are parameters provided for the frequency
range of a respective sub-channel and where x constitutes
e.g. the deviation of the frequency within this frequency
range from the center frequency of this sub-channel.
30

Based on the received parameters, the filter structures
compensate in each signal output by the analysis portion
21 the effects of fading and frequency selectivity in the
respective sub-channel on the radio interface.

After this channel equalization, the real part of the in-
phase component and the quadrature component of a
respective signal are taken at a unit 30, 31 and
5   subjected to a respective slicer (not shown), in order to
obtain the restored 2M sub-channel symbol sequences $\hat{I}_k(m)$,
$\hat{I}_{2M-1-k}(m)$. In the notation $\hat{I}_k(m)$, $\hat{I}_{2M-1-k}(m)$, the indices k,
2M-1-k indicate again the respective sub-channel, while
the parameter m is again a time index.
10

Simulation results indicate that using such a piece-wise
linearly frequency dependent model for the channel
frequency response in channel equalization along with the
proposed equalizer structure, a considerable reduction in
15   the number of sub-channels of up to a factor of about 10
is possible in comparison to the basic OFDM systems.

Compared to the 0th order ASCET of figure 1, the proposed
system has a better performance for a given number of
20   sub-channels, or enables a reduction of sub-channels for
a given performance, since the channel response of a sub-
channel is not assumed to be a constant value. Compared
to known higher-order ASCETs, the proposed system is less
complex, since a simplified model is used for the channel
25   response.

It has to be noted that there are various possibilities
to order the components of the filter structure and the
units taking the real part. The ordering can be done
30   without effecting the overall response. Still, the best
order from the implementation point of view would
probably be to arrange the complex allpass phase
correction stages closest to the analysis portion,
followed by a phase rotation by a complex multiplier.

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combined with taking the real part, i.e. calculating only the real part of the output, and finally an amplitude equalizer for the real signal.

5   While there have shown and described and pointed out fundamental novel features of the invention as applied to a preferred embodiment thereof, it will be understood that various omissions and substitutions and changes in the form and details of the devices and methods described

10 may be made by those skilled in the art without departing from the spirit of the invention. For example, it is expressly intended that all combinations of those elements and/or method steps which perform substantially the same function in substantially the same way to

15 achieve the same results are within the scope of the invention. Moreover, it should be recognized that structures and/or elements and/or method steps shown and/or described in connection with any disclosed form or embodiment of the invention may be incorporated in any

20 other disclosed or described or suggested form or embodiment as a general matter of design choice. It is the intention, therefore, to be limited only as indicated by the scope of the claims appended hereto.

What is claimed is:

5   1.   A method for a filter bank based signal processing
          system, said method comprising:
          - performing a filter-bank based analysis for
             converting a complex higher-rate channel signal
             into oversampled lower-rate sub-channel signals,
10   each sub-channel corresponding to a different
             frequency range; and
          - processing oversampled lower-rate sub-channel
             signals with a polynomial model of a system
             frequency response within the frequency range of
15   the respective sub-channel.

     2.   The method according to claim 1, wherein sine-
          modulated and cosine-modulated filter-bank sections
          are employed for realising said oversampling filter-
20   bank based analysis.

     3.   The method according to claim 1 or 2, wherein said
          analysis is oversampling two times and provides
          output signals in in-phase and quadrature (I/Q)
25   format.

     4.   The method according to one of the preceding claims,
          wherein at least one of said polynomial models of a
          system frequency response within the frequency range
30   of a respective sub-channel is a linear frequency-
          dependent model.

5.   The method according to one of claims 1 to 3, wherein
     each of said polynomial models is of an order between
     1 and 3.

5   6.   The method according to one of the preceding claims,
          wherein at least one of said polynomial models of a
          system frequency response within the frequency range
          of a respective sub-channel is composed of different
          polynomial models of a system frequency response for
10   different sub-frequency ranges.

     7.   The method according to one of the preceding claims,
          wherein at least one of said polynomial models of a
          system frequency response within the frequency range
15   of a respective sub-channel comprises an amplitude
          response model and a phase response model for said
          sub-channel.

     8.   The method according to claim 7, wherein said sub-
20   channel processing is realized with a filter
          structure comprising for each sub-channel at least
          one amplitude equalizer using said amplitude response
          model for the respective sub-channel and an allpass
          filter using said phase response model for the
25   respective sub-channel.

     9.   The method according to claims 7 or 8 , comprising
          for each sub-channel in this order: performing based
          on said phase response model for the respective sub-
30   channel a complex allpass phase correction and a
          phase rotation, in which phase rotation only the real
          part of an output signal is calculated, and applying
          based on said amplitude model for the respective sub-

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channel an amplitude equalization on said output real
signal.

10. The method according to one of the preceding claims,
     employed in a transmultiplexer configuration, in
     which a filter-bank based synthesis is employed for
     converting lower rate sub-channel signals into said
     complex higher-rate channel signals.

11. The method according to claim 10, in which said
     transmultiplexer configuration is used in a channel
     equalization in a filter bank based multicarrier
     system, wherein said sub-channel processing forms
     part of said channel equalization.

12. The method according to one of claims 1 to 9,
     employed in an analysis-synthesis configuration, in
     which a filter-bank based synthesis is employed for
     converting said lower-rate sub-channel signals on
     which said sub-channel processing was performed into
     complex higher-rate channel signals.

13. The method according to claim 12, in which said
     analysis-synthesis configuration is used in a channel
     equalization in a single carrier transmission system,
     wherein said sub-channel processing forms part of
     said channel equalization.

14. A unit for performing a signal processing in a filter
     bank based signal processing system, said unit
     comprising:
     -   an analysis filter-bank with a plurality of sub-
         channel filters for converting a complex higher-
         rate channel signal input to said unit into

oversampled lower-rate sub-channel signals, each
sub-channel corresponding to a different frequency
range; and

- a filter structure for processing oversampled
  5   lower-rate sub-channel signals with a polynomial
           model of a system frequency response within the
           frequency range of the respective sub-channel.

15. The unit according to claim 14, wherein said analysis
10   filter-bank comprises sine-modulated and cosine-
          modulated filter-bank sections for realising said
          oversampling.

16. The unit according to claim 14 or 15, wherein said
15   analysis filter-bank realizes a double oversampling
          and provides output signals in in-phase and
          quadrature (I/Q) format.

17. The unit according to one of claims 14 to 16, wherein
20   said filter structure employs at least one polynomial
          model of a system frequency response within the
          frequency range of a respective sub-channel which is
          a linear frequency-dependent model.

25 18. The unit according to one of claims 14 to 16, wherein
          the respective polynomial model employed by said
          filter structure is of an order between 1 and 3.

19. The unit according to one of claims 14 to 18, wherein
30   said filter structure employs at least one polynomial
          model of a system frequency response within the
          frequency range of a respective sub-channel which is
          composed of different polynomial models of a system

frequency response for different sub-frequency ranges.

20. The unit according to one of claims 14 to 19, wherein said filter structure employs at least one polynomial model of a system frequency response within the frequency range of a respective sub-channel which comprises an amplitude response model and a phase response model for said sub-channel.

21. The unit according to claim 20, wherein said filter structure comprises for each sub-channel at least one amplitude equalizer using said amplitude response model for the respective sub-channel and an allpass filter using said phase response model for the respective sub-channel.

22. The unit according to claim 20 or 21, wherein said filter structure comprises for each sub-channel in the following order: an allpass section filtering received signals based on said phase response model for the respective sub-channel, a phase rotation portion rotating the phase of signals output by said allpass phase equalizer based on said phase response model for the respective sub-channel, which phase rotation portion calculates only the real part of said phase rotated signals, and an amplitude equalizer performing an amplitude equalization on real signals provided by said phase rotation portion based on said amplitude response model for the respective sub-channel.

23. The unit according to one of claims 14 to 22, wherein
    said unit is a receiver for a transmultiplexer
    system.

5   24. The unit according to claim 23, which is used in a
    channel equalization in a filter bank based
    multicarrier system, wherein said filter structure
    performs said sub-channel processing as part of said
    channel equalization.

10

25. The unit according to one of claims 14 to 22, wherein
    said unit is a conversion unit for an analysis-
    synthesis filter bank system.

15 26. The unit according to claim 25, which is used in a
    channel equalization in a single carrier transmission
    system, wherein said filter structure performs said
    sub-channel processing as part of said channel
    equalization.

20

27. A filter bank based signal processing system
    comprising a unit for performing a signal processing
    with:
    -   an analysis filter-bank with a plurality of sub-
25   channel filters for converting a complex higher-
        rate channel signal input to said unit into
        oversampled lower-rate sub-channel signals, each
        sub-channel corresponding to a different frequency
        range; and
30 - a filter structure for processing oversampled
        lower-rate sub-channel signals with a polynomial
        model of a system frequency response within the
        frequency range of the respective sub-channel.

28. The filter bank based signal processing system
    according to claim 27, wherein said unit is a
    receiver and wherein said filter bank based signal
    processing system is a transmultiplexer system
5   further comprising a synthesis filter-bank for
    converting lower-rate sub-channel signals into
    complex higher-rate channel signals for transmission
    to said receiver.

10 29. The filter bank based signal processing system
    according to claim 27, wherein said system is an
    analysis-synthesis filter bank based signal
    processing system further comprising a synthesis
    filter-bank for converting lower-rate sub-channel
15 signals on which said sub-channel processing was
    performed by said unit into complex higher-rate
    channel signals.
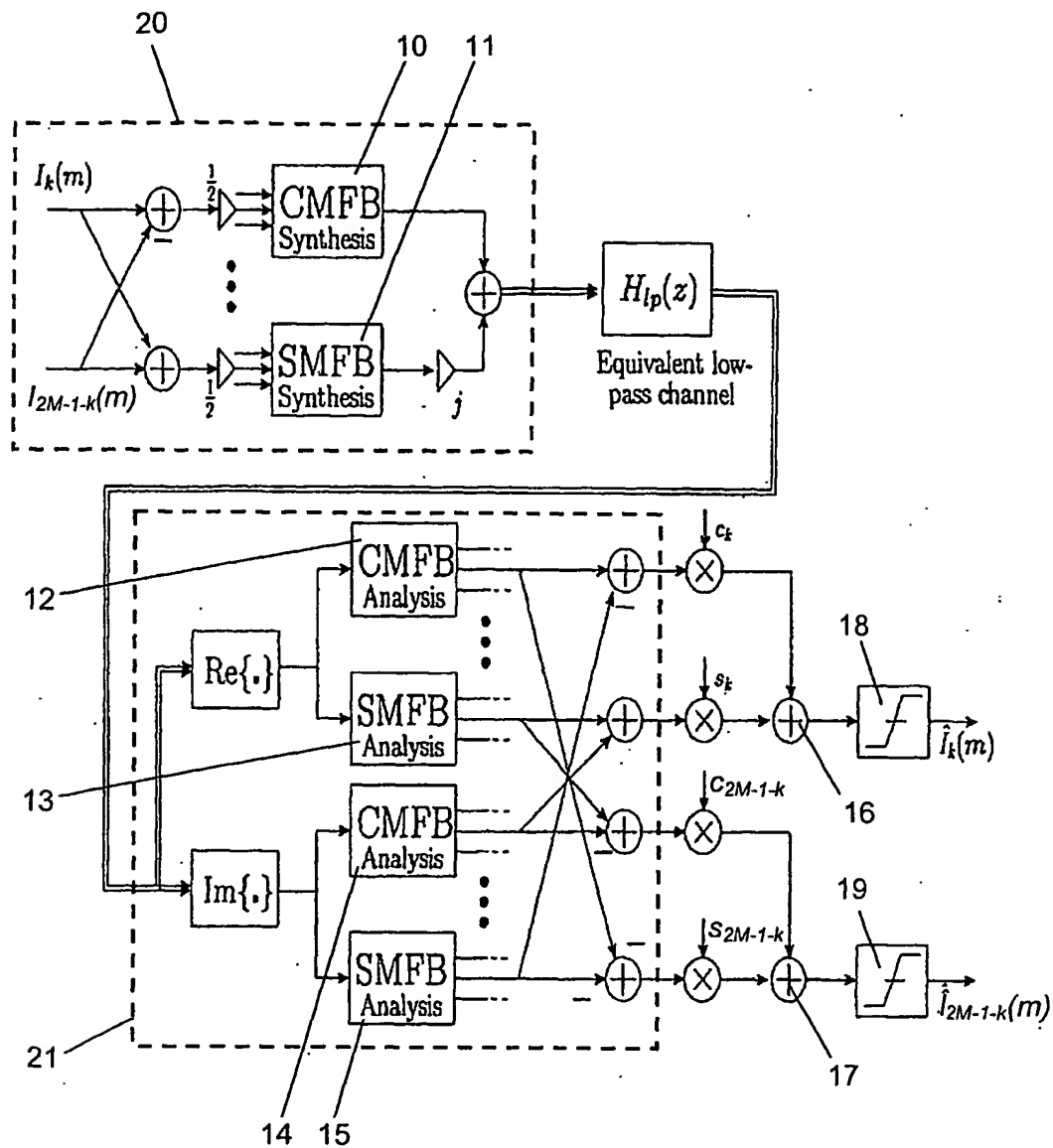
FIG. 1 (Prior Art)
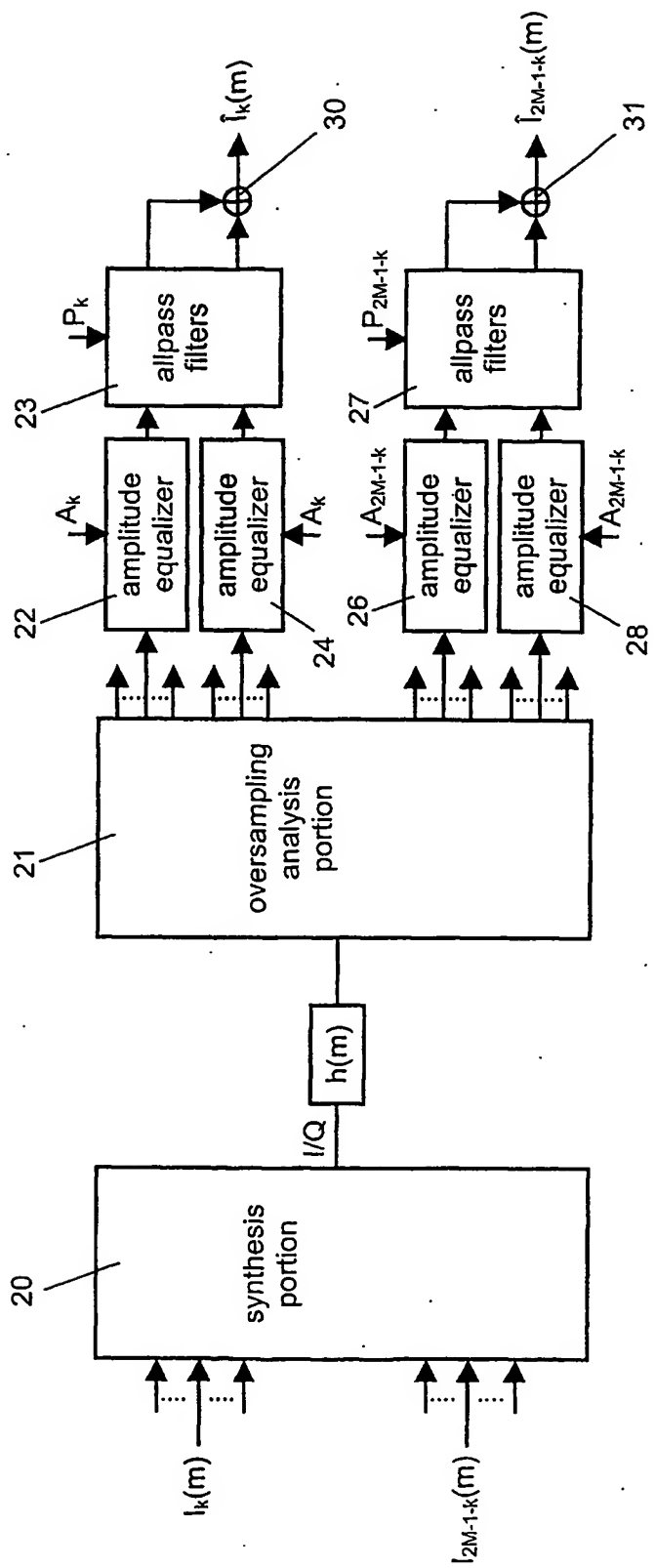
FIG. 2

INTERNATIONAL SEARCH REPORT

PCT/IB 03/06271

A. CLASSIFICATION OF SUBJECT MATTER
IPC 7   H04L27/26   H04L27/28

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED
Minimum documentation searched (classification system followed by classification symbols)
IPC 7   H04L   H04Q

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EPO-Internal, WPI Data, INSPEC

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	MARKKU RENFORS ET AL: "On channel equalization in filter bank based multicarrier access systems" 19/9/1999 IEEE, [Online] XP010352952 ISBN: 0-7803-5435-4 Retrieved from the Internet: <URL:http://ieeexplore.ieee.org/iel5/6454/17258/00797120.pdf?tp=&arnumber=797120&isnumber=17258> [retrieved on 2004-03-17] section 1, 2, 3.3 and 3.4, abstract --- -/--	1-29

[X] Further documents are listed in the continuation of box C.   [X] Patent family members are listed in annex.

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Date of the actual completion of the international search	Date of mailing of the international search report
23 March 2004	13. 04. 2004

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C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT		
Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	ARI VIHOLAINEN ET AL: "Equalization in filter bank based multicarrier systems" 1999 IEEE, [Online] XP002274525 ISBN: 0-7803-5682-9 Retrieved from the Internet: <URL:http://ieeexplore.ieee.org/iel5/6565/17615/00814446.pdf?tp=&arnumber=814446&isnumber=17615> [retrieved on 2004-03-17] section 1 and 3.3, abstract ---	1-29
A	LING QIN ET AL: "Adaptive sub-channel equalization in miltucarrier transmission" 1999 IEEE, [Online] XP002274526 ISBN: 0-8186-7919-0 Retrieved from the Internet: <URL:http://ieeexplore.ieee.org/iel3/4635/13191/00599517.pdf?tp=&arnumber=599517&isnumber=13191> [retrieved on 2004-03-17] sectin 1 and 2, abstract ---	1-29
A	WO 01 47200 A (THOMSON LICENSING SA ;RAMASWAMY KUMAR (US); KNUTSON PAUL GOTHARD () 28 June 2001 (2001-06-28) page 1 -page 6 abstract ---	1-29
A	HELMUT BÖLCSKEI ET AL: "Oversampled cosine modulated filter banks with perfect reconstruction" 1999 IEEE, [Online] XP000848628 Retrieved from the Internet: <URL:http://ieeexplore.ieee.org/iel4/82/15534/00718813.pdf?tp=&arnumber=718813&isnumber=15534> [retrieved on 2004-03-17] Introductionsection, abstract -----	1-29

Patent document cited in search report		Publication date	Patent family member(s)		Publication date
WO 0147200	A	28-06-2001	AU	2110101 A	03-07-2001
			WO	0147200 A1	28-06-2001